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[continued on next page]

(54) Title: METHOD FOR THE REPRODUCTION OF SOUND WAVES USING ULTRASOUND LOUDSPEAKERS

(54) German Title: VERFAHREN ZUR WIEDERGABE VON AUDIOSCHALL MIT ULTRASCHALL-LAUTSPRECHERN

[see source for figure]

SIMPLE EMBODIMENT OF THE ULTRASOUND LOUDSPEAKER

1...LOW FREQUENCY
2...FREQUENCY RESPONSE LINEARIZATION 1
3...TWO-SIDE BAND AMPLITUDE MODULATION, CARRIER IN THE ULTRASOUND RANG
4...ERROR COMPENSATION
5...FREQUENCY RESPONSE LINEARIZATION 2

(57) **Abstract:** [see source for English text]

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Process for the Reproduction of Audio Sound, using ultrasound loudspeakers.

This invention relates to a process for the reproduction of audio sound, with the help of ultrasound loudspeakers as well as a design and construction of the ultrasound loudspeakers.

It is known that a loudspeaker can be assembled from several ultrasound radiators according to J. Acoust. Soc. Am., 73, No. 5, May 1983: "The audio spotlight: An application of nonlinear interaction of sound waves to a new type of loudspeaker design." Using such ultrasound radiators, audio sound can be radiated within a frequency range in which the audio sound itself can no longer be perceived by the human ear. By means of nonlinear effects in the air—at high sound pressure and superposition of two ultrasound waves—one can generate an audible sound. The frequency of the ultrasound is high when compared to the usual audio signals; this means that the radiation of the sound—on account of its small wavelength and the converter dimensions of the ultrasound radiator, which, compared to that, are too great—takes place in a manner that is directed very strongly in terms of space. The frequency dependents of the directional characteristic of conventional loudspeaker—spherical radiators at low frequencies, directional transmitters at high frequencies—hardly enters the picture at all in case of an ultrasound loudspeaker.

Furthermore, a different effect is described in the Conference Volume AES, 26-29 September 1998, San Francisco, California, "The Use of Airborne Ultrasonics for Generating Audible Sound Beams." Here again, considerations are known regarding the generation of an audible sound based on the radiation of audio sound by means of ultrasound.

Furthermore, the phenomenon of the generation of sound waves by means of ultrasound radiators is also known from the magazine Audio, No. 8, 1997, pages 7-8. Here one can find a description to the effect that, by means of a loudspeaker system, a first signal of 200 kHz is radiated and the loudspeaker system radiates a second signal with the same frequency of 200 kHz, with the second signal being modulated with the audio sound signal (20 Hz to 20 kHz). As a result of the nonlinear behavior of the air, in case of superposition of the two signals, a mixed result is generated so that the difference between both signals, from each other, is audible as an acoustic sound.

Reference is made to the following publications by way of additional state of the art: US-A-4,872,148, US-A-4,439,642, US-A-4,439,641, US-A-4,409,441, US-A-4,280,204, US-A-4,199,246,WO-A-85/02748, EP-A-0 164 342, EP-A-0 154 256, CA 1 274 619, CA 1 215 164, CA 1 195 420, CA 1 120 578, AU-A-28287/77, AU-A-510193, WO98/39209, WO98/02976, WO98/02977, WO98/02978, WO98/26405, GB-A-2 225 426, DE-A-27 39 748, US-A-5,375,099, CA 1 274 619, DE-A-196 28 849, US-1,616,639,US-A-1,951,669, US-A-2,461,344, US-A-3,398,810. Other features of the ultrasound loudspeaker are described in the above-mentioned bibliographic references.

Although there have been various attempts at making ultrasound loudspeakers, such a product so far has not been able to prevail on the market. This is due to the fact that, in spite of the special properties of ultrasound loudspeakers, there are indeed some problems that partly are connected to the essence of ultrasound propagation, whereas, on the other hand, they are also sound ultrasound radiator itself.

The object of the invention is to improve a process for the reproduction of audio sound, as well as an ultrasound loudspeaker, compared to past attempts, so that qualitatively high-grade sound reproduction will be possible. This problem is solved according to the invention with the help of a process according to Claim 1 and an ultrasound loudspeaker according to Claim 2. Advantageous developments are described in the subclaims and in the description to follow.

The invention-based process connects low-frequency audio sound with the strong directional characteristic of ultrasound. The directional characteristic of the loudspeaker is thus almost independent of the signal frequency. The following might be pointed out for a better understanding of the invention and its essence: Mathematically, using the formulas of nonlinear acoustics, one can show that, at a high sound pressure level ($p > 110$ dB at 40kHz), due to the nonlinearity of the medium called air, new waves are generated whenever several waves are in a reciprocal relationship to each other. The frequencies of these waves correspond to the sum and the differential frequency of the original waves as well as multiples thereof ($n \blacktriangleright \omega_1 \pm m \blacktriangleright \omega_2$ where ω_1 and ω_2 are frequencies of the initiated soundwaves (tones) and n, m are whole numbers). The sum and differential frequencies occur in each frequency range. Definite advantages, compared to conventional loudspeakers, result in the ultrasound range in that one can implement a very strong directional characteristic of the converters which lies outside human auditory range. The initiating signals—other words the ultrasound waves—are inaudible here.

If, for example, a first tone, with a frequency of 200 kHz, and a second tone, with a frequency of 201 kHz, is radiated at high sound pressure into the air, then sum and differential tones are generated in the superposition zone of the two tones. The first sum tone (f=200kHz+201kHz=401kHz) is not audible. The first differential tone (f=200kHz-201kHz=1kHz) is used to generate audible sound (Figure 4). This differential tone is much louder than all other tones that are generated with respect to the reciprocal interaction. Sum and differential tones are generated first as distortion products in a nonlinear medium, such as air.

The generated differential tones here have the property to the effect that the propagation of the differential tones (secondary sound) takes place in the direction of the ultrasound that is to be generated (initiating tones, primary sound). Furthermore, the differential tones are audible only in the range of ultrasound, that is to say, the directional characteristic of the differential tones corresponds to that of ultrasounds. Finally, the sound pressure of the differential tones rises with the frequency of the ultrasound.

In actually making an invention-based ultrasound loudspeaker, the audio signal, which is to be reproduced and which still has a low frequency, is first of all subjected to frequency response linearization (Figure 1, Figure 2). This signal is then tied in with a carrier signal, in the ultrasound frequency range, by means of a two-sideband amplitude modulation. Then this ultrasound signal is subjected to a dynamic (error compensation (compression)), the compressed signal is subjected to a second frequency response linearization, and this signal is again supplied to the ultrasound loudspeaker.

As an alternative to the previously described formation of the ultrasound signal, one can provide a one-sideband amplitude modulation instead of the two-sideband amplitude modulation; here, the ultrasound carrier is preferably suppressed by several dB, for example, 12dB (Figure 2).

The ideal center frequency, that is to say, the average value between the ultrasound carrier frequency and the sideband frequency (range) of the radiated ultrasound signal results from the intended use. Two major groups can be mentioned here: A. in near range up to about 50 cm; B. use at an interval of more than 50 cm up to remote acoustic irradiation.

From this subdivision of areas, one can derive differing requirements for the center frequency. The level of the audible sound pressure depends decisively on the sound pressure of the ultrasound signal, the nonlinearity parameter of the medium, the frequency of the developing audio signal, as well as the distance to the source and the attenuation of the medium. The differential frequency wave DFW—in other words, the audible sound—is built up with increasing distance to the source. Due to the attenuation of the ultrasound wave in the air, one can reach the highest sound pressure at a certain distance, until the level drops again due to attenuation as a result of growing distance. The attenuation of ultrasound in the air again depends on the ultrasound frequency. The higher the frequency, the higher is also the absorption of ultrasound in the air.

Here is what that means for practical uses: An ideal frequency range of about 40 kHz up to 500 kHz (or more) can be given for uses at an interval of more than 50 cm up to several meters. The frequency range, on one hand, is chosen high enough in order, in the most effective possible way, to generate a differential frequency wave and to ensure an adequate frequency interval to the audible sound, although, on the other hand, it should be low enough so that the attenuation due to the air does not exert too great influence on the audio sounds. Another criterion is the directional characteristic of the ultrasound radiator. The higher the radiator frequency, the more directional will the radiation be.

A higher frequency makes good sense for the near range (less than 50 cm) because the absorption of the air is negligible in the near range, whereas the dimensions of the ultrasound converter, depending on the use, are so small that a stronger directional effect is not achieved as a result of the shape of the converter but can be implemented only by raising the ultrasound frequency.

The frequency shift of the low-frequency signal (speech, music, noises, sounds) in the ultrasound range takes place by means of amplitude modulation. This generates a carrier signals as well as an upper and a lower sideband which contain the modulated information. At high sound pressure, the carrier signal, for example, 200 kHz, and the lower sideband, are radiated via a converter and are superposed in the air. As a result of the nonlinear behavior of the air, a signal is generated here whose frequency corresponds to difference from the carrier frequency and the sideband frequency. The higher the frequencies of the radiated tones, at constant amplitude, the louder are the developing differential tones. The sound pressure of the differential tones rises quadratically with the differential frequency of the radiated ultrasound tones. By means of a high ultrasound frequency, one can maximize the achievable directional effect and one can enlarge the frequency interval of the radiated ultrasound to the human auditory range.

The sound pressure of the differential frequencies results, among other things, from the product of the signals that are to be mixed. The carrier is radiated at its full level during the radiation of an amplitude-modulated signal, even in case of a modulation pause, that is to say, when no signal is applied to the modulator. The amplitude of the carrier signifies a constant noise annoyance for the ears and a permanent electrical charge of the converters. The amplitude of a sideband is $m x A_T/2$ (where m=modulation index and A_T=carrier amplitude in case of an ordinary amplitude modulation). The carrier is constantly radiated and has a greater amplitude than the sideband which is modulated at the cycle of the low frequency. These above-mentioned problems can be meaningfully solved with the measures described below. One can achieve a noise reduction if the amplitude of the carrier is reduced, for example, by a filter or already in the modulator by partial carrier suppression and if the amplitude of the upper sideband is raised simultaneously. As a result, the constant level is reduced and the relative change of the level, related to the carrier, becomes greater due to modulation. In case of carrier suppression, the lower sideband must be strongly suppressed in order to prevent the two sidebands from becoming intermingled with each other—something that would cause severe distortions. The above-described measure can also be generally referred to as "carrier reduction."

If the carrier amplitude is modulated with the amplitude of the signal that is to be transmitted, then no signal is radiated in case of a pause in modulation. This requires and additionally controlled compressor stage which will balance out the amplitude errors that result from the modulation of the carrier. To eliminate the above-described problem, one can thus perform a modulation of the carrier amplitude at the cycle of the signal to be modulated.

Furthermore, one can cope with one of the above-described problems in that one achieves a compression of the signal to be modulated so that the signal's dynamics will be reduced and so that, as a result, in particular, the loudness of the soft signal passages will be raised. In that way, one can select the modulator in an optimum fashion. After modulation, compression must be adjusted again by expansion in order to get the original dynamics. Very good results were achieved with the described compression of the modulation signal prior to modulation.

Another measure to eliminate the above-mentioned problem consists of the following: During modulation pauses, the triggering of the converters with the carrier signal is suppressed (mute circuit) so that the modulator output signal is masked when no input signal is applied.

The amplitude-modulated low-frequency oscillation is radiated at high sound pressure with a converter. Due to the reciprocal interaction between the carrier oscillation and the modulated sideband, in the air, there is generated a differentiated frequency spectrum that corresponds to the spectrum of the low frequency. A one-sideband modulation is suitable in a particularly preferred manner in order to achieve a low nonlinear distortion factor. If the carrier is partially suppressed in an ordinary two-sideband amplitude modulation, then the suppression of the lower sideband is inevitable because the mixing of the two sidebands with each other brings about additional differential frequencies that make themselves noted rather undesirably in the form a nonlinear distortion factor.

When one uses piezoelectric converters, the radiation of the modulated signal however is so narrow-band that the lower sideband is reproduced only very softly. As a result, the mixing of the two sidebands is negligible in terms of sound pressure. But this presupposes that the carrier is so loud that the mixing of carrier and sideband results in a signal that is much louder than the mixing of the sidebands with each other. Accordingly, the modulation is accomplished either as an ordinary two-sideband amplitude modulation or as one-sideband amplitude modulation, where the carrier is suppressed for further functional optimization, for example, by 12dB.

The connection between the electrical input signal of the piezoelectric converters and the sound pressure level of the differential tones is not linear. One can achieve a linear transmission (dynamics compression) with a compensation circuit.

Using a frequency response linearization—which is required especially in case of piezoelectric converters with strong nonlinear frequency response—frequency—dependent amplitude errors of the transmission system are adjusted. The rectification can be done prior to modulation in the low-frequency range or after modulation, in the ultrasound range. Rectification after modulation offers the advantage that, as a result, the selection reserve of the modulator is not restricted when a frequency range is raised. The differential tone wave is generated in the radiated ultrasound cone. The cross-section of the cone here influences the resultant audio-frequency response. The audible signal is generated at a boundary surface that is held into the sound beam. The lower boundary frequency here depends on the cross-section surface that is brought into the beam. Rectification attuned to the surface of the reflector (surface-related rectification) is needed in order to achieve a linear frequency response for a reflector on a wall.

The maximum of the sound pressure results at a certain distance from the ultrasound source. It occurs at varying intervals for differing audio frequencies. A linear frequency response therefore can develop for a certain distance only as a result of an especially

distance-related rectification. Signal processing therefore must contain a special, distance-dependent frequency response rectification for a linear frequency response.

A larger number of converters are parallel-connected to generate a high ultrasound level. In the process, it was discovered that the arrangement of the converters plays a big role. For example, converters are arranged on a plate as tightly together as possible so that the depth reproduction of the loudspeaker will be softer than with the same number of converters in a ring-shaped pattern.

The described analog amplitude modulation can also be done digitally. Here, there is a possibility for the multiplication of a sinusoidal oscillation (carrier) with a low-frequency signal, partial suppression of the carrier, as well as suppression of the lower sideband, with a digital signal processor building lock. The frequency response contours can also be done relatively easily when one uses a digital signal processor.

The level of the audio sound pressure among other things also depends on the nonlinearity parameter of the acoustically permeable medium. For air, the parameter amounts to _=1.2. For the medium water, it is _=3.5. It was now found that, in case of a water-air bubble mixture, one can give an extreme value of amounting to more than 5,000; this means that, compared to the medium air, with a water/air mixture, the sound pressure can be increased by a factor of 4,000. In this way, it is possible, for example, to create a water/air mixture in an earphone earpiece so that the water/air mixture medium will be arranged between the ultrasound radiator and the listener and so that the sound pressure of the audio signal will be increased.

The audio sound pressure however can also be further enlarged by other means: Due to the increasing steepness of the wave front in the coarse of propagation, something that is equivalent to the origin of upper harmonic waves. According to an energy balance, the energy that is located in the upper harmonic waves, is not available for the differential tone waves. There is, so to speak, an energy flow from the ground [carrier] wave to the upper harmonic waves. If it is possible to slow this energy flow down, then one could increase the audio sound pressure. Here is one proposal for doing this:

A sound-permeable medium contains small cavities that, together with the material, result in a plurality of Helmholz-Resonators. The Resonators are attuned to the first upper harmonic wave of the signal and thus brake the energy flow to the higher upper harmonic waves. If one fills the cavities with a nonlinear medium, for example, a liquid, then, by means of this measure, one can achieve a higher value for the nonlinearity parameters, as a result of which the sound pressure of the differential tones is increased.

With this technology, one can build reflectors that amplify the sound pressure of the differential tones in a passive manner.

By means of the described "attenuation plate," one can achieve a higher audio sound, with simultaneously reduced ultrasound, for an ultrasound loudspeaker built into the headrest of a car. For cable-less earphones, it might be conceivable to transmit in

inaudible ultrasound in a wire less fashion and to move the differential tones to an adequate level via the above-described absorber.

Using the formula of nonlinear acoustics, one can show mathematically that new waves are generated, at a high sound pressure level (p>110dB at 40kHz), due to the nonlinearity of the medium of air, when several waves are in a reciprocal direction [changing direction] with respect to each other.

The frequencies of these waves correspond to the sum and differential frequencies of the original waves as well as a multiple thereof.

($n * \omega_1 \pm m\omega_2$ with ω_1, ω_2: frequencies of the initiated tones and n, m: gaanic numbers.)

The sum and differential frequencies occur in each frequency range. Definite advantages, compared to conventional loudspeakers, result in the ultrasound range in which one can achieve a very strong directional characteristic of the converters and which lies outside the human auditory range; the initiating signals here are inaudible.

Example:

If a tone with frequency 200kHz and a second tone with frequency 201kHz is radiated into the air at high sound pressure, then sum and differential tones are generated in the superposition zone of the two tones. The first sum tone (f=200kHz+201kHz=401kHz) is not audible. The first differential tone (f=200kHz-201kHz=1kHz) is used to generate audible sounds. It is furthermore much louder than all other tones generated during this reciprocal interaction. Distortion products — that yield the sum and differential tones — are generated only in a nonlinear medium such as air.

Properties of Generated Differential Tones

The secondary sound (of the differential tones) is propagated in the direction of the primary sound (of the initiating tones).

The secondary sound is audible only in the range of the primary sound, that is to say, the directional characteristic of the secondary sound corresponds to that of the primary sound.

The sound pressure of the differential tones rises with the frequency of the initiating tones.

Technical Implementation (Exemplary Embodiment of Invention)

Figure 1 and Figure 2 show block diagrams of an ultrasound loudspeaker, with Figure 2 representing an improved circuit when compared to Figure 1.

As one can see in Figure 1, the low-frequency audio signal is first of all subjected to a frequency response linearization and then to a two-sideband amplitude modulation (and/or a frequency and/or phase modulation), with the carrier frequency lying in the ultrasound range. Thereafter, one performs a dynamics compression or dynamics error compensation (signal-dependent). This is followed once again by another frequency response linearization and the signal, then output, is supplied to the ultrasound converter.

The circuit according to Figure 2 differs from Figure 1 essentially by virtue of the fact that a one-sideband amplitude modulation is performed instead of the two-sideband amplitude modulation, with the carrier being suppressed by about 12dB in the ultrasound range.

The ideals center frequency, that is to say, the average between carrier frequency and sideband frequency (range) of the radiated ultrasound signal results from the intended use. One can mention two groups:

1. Use in near range up to about 50cm.
2. Use in a range of >50cm and remote acoustic irradiation.

The differing requirements for the center frequency can be derived from this subdivision of ranges. The level of the audible sound pressure depends on the sound pressure of the ultrasound signal, the nonlinearity parameter of the medium, the frequency of the developing audio signal, as well as the distance to the source and the attenuation of the medium. The differential frequency wave is built up with growing distance from the source. Due to the attenuation of the ultrasound wave in the air, one can achieve the greatest sound pressure at a certain distance, until the level drops again as the distance grows due to attenuation. The attenuation of ultrasound in the air again depends on the frequency. The higher the frequency, the higher is also the absorption of sound in the air.

Here is what that means for practical uses: An ideal frequency range of about 80 kHz up to 180 kHz can be given for uses at an interval of more than 50 cm up to several meters. The frequency range, on the one hand, is chosen high enough in order, in the most effective possible way, to generate a differential frequency wave and to ensure an adequate frequency interval to the audible sound, although, on the other hand, it should be low enough so that the attenuation due to the air does not exert too great influence on the audio sounds. Another criterion is the directional characteristic of the radiator. The higher the radiator frequency, the more directional will the radiation be.

A higher frequency makes good sense for the near range because the absorption of the air is negligible in the near range, whereas the dimensions of the ultrasound converter, depending on the use, are so small that a stronger directional effect is not achieved as a result of the shape of the converter but can be implemented only by raising the ultrasound frequency.

Frequency shift of low-frequency signal

The frequency shift of the low-frequency signal (speech, music, noises, sounds) in the ultrasound range takes place by means of amplitude modulation. This generates a carrier signals as well as an upper and a lower sideband which contain the modulated information.

At high sound pressure, the carrier signal, for example, 200 kHz, and the lower sideband, are radiated via a converter and are superposed in the air. As a result of the nonlinear behavior of the air, a signal is generated here whose frequency corresponds to difference from the carrier frequency and the sideband frequency. The higher the frequencies of the radiated tones, at constant amplitude, the louder are the developing differential tones. The sound pressure of the differential tones rises quadratically with the differential frequency of the radiated ultrasound tones. By means of a high ultrasound frequency, one can maximize the achievable directional effect and one can enlarge the frequency interval of the radiated ultrasound to the human auditory range.

Here is an inadequacy connected with amplitude modulation: permanent carrier amplitude. The sound pressure of the differential frequencies results, among other things, from the product of the signals that are to be mixed. The carrier is radiated at its full level during the radiation of an amplitude-modulated signal, even in case of a modulation pause, that is to say, when no signal is applied to the modulator. The amplitude of the carrier signifies a constant noise annoyance for the ears and a permanent electrical charge of the converters. The amplitude of a sideband is $m*A_T/2$ (where m=modulation index and A_T=carrier amplitude). The carrier is constantly radiated and has a greater amplitude than the sideband which is modulated at the cycle of the low frequency. The following measure therefore make good sense:

Carrier Reduction

These above-mentioned problems can be meaningfully solved with the measures described below. One can achieve a noise reduction if the amplitude of the carrier is reduced, for example, by a filter or already in the modulator by partial carrier suppression and if the amplitude of the upper sideband is raised simultaneously. As a result, the constant level is reduced and the relative change of the level, related to the carrier, becomes greater due to modulation. In case of carrier suppression, the lower sideband must be strongly suppressed in order to prevent the two sidebands from becoming intermingled with each other—something that would cause severe distortions.

Modulation of Carrier Amplitude at Cycle of Signal to be Modulated

If the carrier amplitude is modulated with the amplitude of the signal that is to be transmitted, then no signal is radiated in case of a pause in modulation. This requires and additionally controlled compressor stage which will balance out the amplitude errors that result from the modulation of the carrier.

Compression of Modulation Signal prior to Modulation

By compressing the signal to be modulated, one can see to it that the dynamics of the signal will be reduced and that, as a result, especially the loudness of the soft signal passages will be raised. As a result, one can select the modulator in an optimum fashion. After modulation, compression must be balanced out again by means of expansion in order to get the original dynamics.

Mute Circuit

The modulator output signal is masked, if no input signal applies, in order to suppress a triggering of the converters with a carrier signal during modulation pauses.

Practical Design of Modulator

When one uses piezoelectric converters, the radiation of the modulated signal however is so narrow-band that the lower sideband is reproduced only very softly. As a result, the mixing of the two sidebands is negligible in terms of sound pressure. But this presupposes that the carrier is so loud that the mixing of carrier and sideband results in a signal that is much louder than the mixing of the sidebands with each other.

Accordingly, the modulation is accomplished either as an ordinary two-sideband amplitude modulation or as one-sideband amplitude modulation, where the carrier is suppressed for further functional optimization, for example, by 12dB.

Dynamics Compression (Dynamic Error Compensation)

The connection between the electrical input signal of the piezoelectric converters and the sound pressure level of the differential tones is not linear. One can achieve a linear transmission with a compensation circuit.

Linearization of the Frequency Response

Using a frequency response linearization—which is required especially in case of piezoelectric converters with strong nonlinear frequency response—frequency-dependent amplitude errors of the transmission system are adjusted. The rectification can be done prior to modulation in the low-frequency range or after modulation, in the ultrasound range. Rectification after modulation offers the advantage that, as a result, the selection reserve of the modulator is not restricted when a frequency range is raised.

Surface-Related Rectification

The differential tone wave is generated in the radiated ultrasound cone. The cross-section of the cone here influences the resultant audio-frequency response. The audible signal is generated at a boundary surface that is held into the sound beam. The lower boundary frequency here depends on the cross-section surface that is brought into the beam.

Rectification attuned to the surface of the reflector (surface-related rectification) is needed in order to achieve a linear frequency response for a reflector on a wall.

Distance-Related Rectification

The maximum of the sound pressure results at a certain distance from the ultrasound source. It occurs at varying intervals for differing audio frequencies. A linear frequency response therefore can develop for a certain distance only as a result of an especially distance-related rectification. Signal processing therefore must contain a special, distance-dependent frequency response rectification for a linear frequency response.

Raising the Sound Pressure by Means of a Large Number of Converters

A larger number of converters are connected parallel to generate the high ultrasound level.

The arrangement of the converters plays a role here: If the converters are arranged as tightly together on a plate as possible, then the depth reproduction of the loudspeaker is softer than in an arrangement where the same number of converters are arranged in a ring-shaped fashion.

Modulation by Digital Signal Processing

The described analog amplitude modulation can also be performed digitally. There is a possibility for the multiplication of a sinusoidal oscillation (carrier) with a low-frequency signal, partial suppression of carrier, as well as suppression of lower sideband with a digital signal processor building block (Figure 3). Frequency response corrections can also be made in a relatively simple manner.

Nonlinearity Parameter

The level of the audio sound pressure depends among other things on the nonlinearity of parameter of the medium. For air, the parameter is $_{\epsilon}$=1.2. For the medium of water, it is $_{\epsilon}$=3.5; for water with air bubbles, one can give an extreme value of $_{\epsilon}$=5,000. Compared to the medium of air, one can thus theoretically achieve a sound pressure that will be higher by a factor of 4,000.

A suitable medium between ultrasound radiator and listener can increase the sound pressure of the audio signal.

The audio sound pressure however can also be further enlarged by other means: Due to the increasing steepness of the wave front in the coarse of propagation, something that is equivalent to the origin of upper harmonic waves. According to an energy balance, the energy that is located in the upper harmonic waves, is not available for the differential tone waves. There is, so to speak, an energy flow from the ground [carrier] wave to the

upper harmonic waves. If it is possible to slow this energy flow down, then one could increase the audio sound pressure.

Here is one proposal for doing this:

A sound-permeable medium contains small cavities that, together with the material, result in a plurality of Helmholz-resonators. The resonators are attuned to the first upper harmonic wave of the signal and thus brake the energy flow to the higher upper harmonic waves. If one fills the cavities with a nonlinear medium, for example, a liquid, then, by means of this measure, one can achieve a higher value for the nonlinearity parameters, as a result of which the sound pressure of the differential tones is increased.

With this technology, one can build reflectors that amplify the sound pressure of the differential tones in a passive manner.

By means of the described "attenuation plate," one can achieve a higher audio sound, with simultaneously reduced ultrasound, for an ultrasound loudspeaker built into the headrest of a car.

For cable-less earphones, it might be conceivable to transmit in inaudible sound ultrasound in a wire less fashion and to move the differential tones to an adequate level via the above-described absorber.

Practical Uses

The audible sound is generated only in the transition zone of the mixed ultrasound signals; therefore, due to the spatially separated radiation of carrier signal and sideband signal over their own converters, one can accomplish an almost point-shaped "projection" of the sound. The irradiation of both signals over a single converter or a converter array on the other hand changes the point-shape characteristic into a linear characteristic along the direction of propagation of the ultrasound.

Practical uses of the ultrasound loudspeaker primarily involve those where the strong directional effect of the loudspeaker is used. Looking at uses (a)-(e), an absorbing material behind the area to be exposed to sonic waves, ensures the prevention of a backward reflection of the ultrasound.

(a) Art objects that "talk" the exposure to sonic waves of an art object in such a way that the sound is audible only in the immediate surroundings of the object. The converter can, for instance, be arranged above the object and will now be audible only within a small area around the object. This does not result in the exposure of the surrounding area to sonic waves.

(b) Active noise compensation, for car, aircraft, bus, train: The surrounding noises are picked up and analyzed by means of a microphone. A signal with opposite phase is generated with the help of an electronic circuit and is radiated by way of the

ultrasound transmission method in a manner coordinated with the seat and in a directed manner. The superposition of the sound with the generated countersound brings about a reduction in the surrounding noises.

(c) Conference systems for spatially addressable exposure to sonic waves in various languages: The individual seats are selectively exposed to sonic waves in conference rooms without the particular neighbor being disturbed. In this way, one can transmit different languages simultaneously and without earphones.

(d) Loudspeaker in aircraft, bus, train as earphone substitute: The strong directional effect of the ultrasound loudspeaker facilitates exposure to sonic waves with loudspeakers instead of earphones. This can be done by making electrically or mechanically swingable radiators which permits "audio demand."

(e) Directed exposure to sonic waves on the stage (prompter).

(f) In the car, as addressable loudspeaker (converters, attached in the roof canopy or the headrest can be controlled via an operating field with matrix display).

(g) Exposure to sonic waves of computer work stations on monitor. Converters are attached around the picture tube of the monitor. The sound is thus directly audible only in front of the monitor.

(h) "Ultrasound wallpaper" or ultrasound ceiling for active noise compensation at home, for function, see above.

(i) Surround loudspeaker: Use of wall reflections: "Projection" of surround information on the walls of the room on which virtual sounds sources are supposed to be located. The rear boxes thus need not necessarily be set up behind the listener.

(j) Exposure to sonic waves in connection with public address system uses: Acoustic "illumination" of very specific zones. This calls for the delimitation of the surrounding areas (audio on demand).

(k) No-hands device (in car, for making telephone calls): Due to the strong directional effect of the loudspeaker, one can make sure—assuming the microphone is suitably attached—that there will be no acoustic feedback between the loudspeaker sound and the picked up microphone sound. Combinations of ultrasound loudspeaker and directional microphone to avoid acoustic feedback: The loudspeaker, for instance, is arranged over the listener while the directional microphone is aimed frontally at the speaker. The powerfully directed sound of the ultrasound loudspeaker does not reach the microphone so that there cannot be any acoustic feedback (for example, in TV studios when there are questions from the audience).

(l) If an ultrasound loudspeaker is installed at each seat, then one can relay a phone call to every seat, without any need for passing the phone receiver on and on.

Inaudible ultrasound is radiated into the air, exclusively, via a special converter, during the process for the reproduction of audio sound, described here.

An audible sound is generated, at high sound pressure and with the superposition of two ultrasound waves due to nonlinear effects in the air. The ultrasound has a high frequency, compared to the usual audio signals; as a result, the radiation of the sound takes place in a powerfully spatially directed manner on account of its short wavelength and the comparatively relatively large converter dimensions. The frequency dependents of the directional characteristic of conventional loudspeakers (spherical radiators at low frequencies, directional radiators at high frequencies) hardly occurs in this loudspeaker.

The process combines low-frequency audio sound with the strong directional characteristic of ultrasound. The directional characteristic of the loudspeaker thus is almost independent of the signal frequency.

Reduction of Distortions during Amplitude Modulation

The modulated signal is radiated with ultrasound converters. If the signal is a two-side-modulated amplitude-modulated signal, then one can reduce the distortions, caused as a matter of principle, in the following manner:

1. By narrow-band converters with high-grade quality,
2. In wide-band converters, by a filter connected in series before.

The filter is eliminated in narrow-band converters because the transmission function of the converters is already equivalent to that of a narrow-band filter.

The system is to be so attuned that the carrier frequency will come to rest roughly at the 6dB point of the filter flank. Cutting the lower sideband off brings about a reduction in distortions.

Temperature-dependent drift of the filter flank of narrow-band converters and filters must be compensated for by follow-up of carrier frequency. The carrier frequency follow-up is done if at all possible during signal pauses.

To increase the comprehension of speech, signal filtering of the audio signal to be modulated should be performed in connection with speech reproduction. The filter is to be so designed that there will be an attenuation of 3dB/O, starting with a signal frequency of 1kHz.

Reduction of Distortions due to Converter Geometry

If the dimensions of the converter exceed a figure of about ¼ of the lowest low-frequency wavelength to be radiated, then, there will—in the near field of the converter—be increasing distortions due to the differences in the running times of the signals. The

dimensions of the converter therefore should not be made smaller than the mentioned wavelength.

Supplement for Technical Conversion of Modulation

An even more forcefully directed radiation of the audio band can be achieved in the following manner:

The sound pressure of the audio band depends on the product of the sound pressures of the carrier sound and of the sound band. By raising the sound pressure—either of the carrier or of the sideband—there will be an increase in the resultant sound pressure in the audio frequency range. The radiation of a wide frequency range at high sound pressure will cause certain difficulties.

The radiation of carrier and sideband via a converter or a group of converters establishes stiff requirements for the converters. The audiowave is generated in the entire superposition area of the signals due to almost identical radiation conditions of carrier and sideband. This leads to a relatively wide radiation. One can achieve an even sharper directional effect by radiating carrier and sideband via separate converters.

A special, very narrow-band, sensitive and very strongly directional converter generates the carrier signal, while the sideband is superposed by a wider-band converter/converter array. The audio sound pressure results from the product of the two ultrasound sound pressures that are to be superposed; therefore, via the sound pressure of the carrier, one can adjust the sound pressure of the audio wave within wide limits and at the same time, the level of the ultrasound carrier can be reduced when the loudness of the level is set low. The superposition of the sound waves and the generation of mixed products however takes place only in the area where both sound waves fill the space or room equally. This makes for a very pronounced directional effect also for the audiowave by virtue of the very strong possible directional characteristic of the carrier radiator.

Absorption of Ultrasound Signal by Ultrasound Filter

To generate the audio signal from the modulated ultrasound signal, one needs a certain path along which the wave is demodulated in the air. If the ultrasound has covered the required distance, then a filter—that is permeable for audio frequencies but that is impermeable for ultrasound—means that the audiowave is well audible but that the ultrasound signal is heavily attenuated. The filter has no significant effect on the directional characteristic of the converter.

The filter must be so constituted that it will heavily attenuate frequencies above the auditory range, whereas audio frequencies experience only low attenuation. It is most meaningfully arranged at the end of the generation zone.

A long generation zone is required for low audio frequencies; therefore, one can vary the lower boundary sequence of the audio signal by varying the interval between converter and absorber.

Enrichment of Acoustic Pattern by Means of Psychoacoustic Effects

The lower the frequency of the audiowave demodulated in the air, the less will be the sound pressure of the wave, related to constant sound pressure of the ultrasound waves. This is why low frequencies can be reproduced only very softly for physical reasons.

To generate the subjective impression of reproducing low tones, that are objectively not present at all, one can, by means of signal processing, generate a certain upper harmonic tone spectrum that allows this impression to be created. This requires a pre-distortion of the audio signal. The modulator contains a circuit that performs this function.

Other Uses

Virtual Loudspeaker

To allow a sound object seemingly to migrate in space, one must move the loudspeaker in the space by means of conventional loudspeaker technique. This effect can be achieved more effectively with the ultrasound loudspeaker.

By using the reflecting properties for ultrasound, as provided by reverberant surfaces, one can make sure that the reflection of the ultrasound loudspeaker against a wall, or the like, will be perceived in manner similar to a light beam reflected by a mirror so that a virtual source is generated.

There are two possible ways here of doing this:

1. Ultrasound loudspeaker suspended in a rotatable and swingable manner.

2. Ultrasound loudspeaker suspended in a fixed manner, radiating on a movably mounted reflector.

Spatial Signal Carrying with following Converter

When the listener is moving, for example, on a moving sidewalk, escalator, etc, the audio signal can be brought along by swinging the converter so that only the moving listener will be exposed to sonic waves, whereas the surrounding area will not be thus exposed.

The audio sound can also be moved along by switching in ultrasound beams that are above the listener—and that are synchronized with the running speed of the moving sidewalk or the escalator—which beams will always only expose to sonic waves that portion of the area in which the listener just happens to be.

Signal following in Space by means of Phased Array.

The specifically targeted triggering of individual converter elements of an array facilitates special signal following (considering the strong directional characteristic of the ultrasound loudspeaker) without in the process moving the ultrasound radiator. The process involves a combination from the technique of "phased array" and the above-described "ultrasound loudspeaker."

Figures 4a and 4b show the propagation of an audio soundwave that is generated by an ultrasound converter. Here, the ultrasound radiator (ultrasound converter) for example simultaneously radiates the frequencies f_1=101kHz and f_2=100kHz. Resembling a (nonlinear) mixing stage of an AM-center-wave receiver, the mixing products f_1+f_2=201kHz and f_1-f_2=1kHz and their multiples are now generated in the ultrasound beam in the air. The sum frequency f_1 and f_2=201kHz cannot be heard by the human individual but the differential frequency f_1-f_2+1kHz is audible to humans. One can now easily imagine that one modulates f_1 with the audio frequency range Δf=100...200kHz to f_1=100kHz+Δf. The audio frequency 100Hz to 20Hz is then precisely generated in the ultrasound beam as a result of the mixing on the nonlinearity of the air, among other things; this latter audio frequency has a similarly strong directionality, that is, similar to the one that is given in advance by the ultrasound beam.

The virtual audio sound sources (virtual loudspeakers) are generated in the mixing zone of the ultrasound beam; they are added up in the direction of the continuing ultrasound because ultrasound and audiosound are propagated with the same speed of sound (340m/sec). One can visualize this effect with the help of a model. Small loudspeakers are mounted close to each other on a strip; all can radiate audio sound as spherical radiators (Figure 5) and they are triggered with time delay, using the same audio signal. The time delay t between two loudspeakers is so chosen that is will exactly correspond to the time the soundwave needs from one to the next loudspeaker. It can be determined by the relationship $t=c/l_L$ (c=speed of sound). The sound issuing from the first loudspeaker is amplified by the second one, and so forth and so on. The audiosound is therefore very intensely focused by virtue of the plurality of loudspeakers (quasi-infinite many virtual sound sources are generated in the ultrasound beam) that are connected in, along with the running time of the sound, in a site-dependent manner.

In the invention-based ultrasound beam, the audio sound is generated in the ultrasound beam itself. In contrast to radiation by a conventional loudspeaker, it initially becomes louder with increasing distance, until the ultrasound level has decreased to such an extent that the nonlinear effect of the air is no longer in action and so that no further parts are added to the generation of the audio sound. The length of the active zone of audio sound generation in the ultrasound beam will determine the lower boundary frequency of the directed audio sound source. There must be at least so many virtual sound sources that the active zone will have a length of several wavelength at the lower boundary frequency. This is why audio frequencies of less than 100Hz will require a long distance between the listener and the ultrasound radiator (and thus also a high output). The use of

psychoacoustic signal processing, as described earlier, will offer an approach to a solution here.

It follows from the two effects described here that the level and the lower reproduction frequency of the audio signal are site-dependent. The ultrasound level, that basically must be necessarily high to generate the audio sound, may be present only in the active zone of the ultrasound beam. Once the directed audio sound beam has been generated, one can eliminate the ultrasound share with the help of an acoustic low-pass filter (audio sound-permeable ultrasound absorber).

Figures 6a and 6b show typical examples of practical use of the ultrasound radiator which is arranged under a ceiling and that directs ultrasound beams, modulated with audio signals at a wall, from which an ultrasound-absorbing coating (ultrasound reflection coating) is so aligned that ultrasound is absorbed. The audio signals that are then reflected are free of ultrasound and can be heard by an individual in front of the wall.

A customary ultrasound converter can be used for the ultrasound converters. Particularly suitable however are also ultrasound foil converters that—in the manner of a condenser (elektret) converter—have a foil and a correspondingly fashioned counterelectrode (with grooves or perforations).

Another advantageous embodiment variant can be described as follows: Using an instrument for measuring the distance to an ultrasound measurement instrument, one can determine where a listener who is to be exposed to sonic waves happens to be located. If the listener is in a critical area of the ultrasound beam which could damage health, then the ultrasound reproduction is turned off so that the particular person (or animal) will not be exposed to excessively high ultrasound levels. If the ultrasound is to be aimed at a certain area and if this area is still also moving (this, for example, is the case with an individual listener who moves on a stage and who is to be exposed to sonic waves), then it is a good idea here if a device is made by means of which the listener, who is to be exposed to soundwaves, can be located in current terms, so that the exposure sonic waves will then take place preferably only upon the specifically localized area. That can be done, for instance, by having the listener—who is to be exposed to sonic waves—carry with him a transmission device with navigation (for example, GPS) so that he will constantly transmit his own navigation data to a reception device which, in turn, is used to control the swinging of the ultrasound beam. The listener, who is to be exposed to sonic waves, can also be equipped with a so-called TAG-identifier whose precise position is determined by a corresponding interrogator (interrogating unit for the TAG), by means of which, in turn, the swing of the ultrasound beams is then controlled. But all other technical possibilities for the localization of an individual area or several areas can be used in order to control the swinging of an ultrasound beam so that the audio reproduction can be heard always only in the desired narrow area, but not outside the desired area.

Such uses are particularly advantageous in a theater (for the prompter) or in television studio during a television show if the moderator, moving along the stage, is to get instructions that are not to be audible to the public.

The ultrasound beam can be swung with the help of the different techniques in this application, in other words, by swinging the ultrasound beams or by means of a swingable reflector or by so-called "phased array" control, where the ultrasound beams are determined in a directional-electronic fashion.

CLAIMS

1. Process and device for the reproduction of audio sound by means of an ultrasound-generating installation, where the audio signal to be reproduced is tied in, by means of a sideband amplitude modulation, with a carrier signal, in the ultrasound frequency range, where means are provided for subjecting the modulated ultrasound signal to a dynamics error compensation and where the compensated ultrasound signal is possibly subjected to a frequency response linearization and is then supplied to an ultrasound converter (loudspeaker), means being provided to reduce the amplitude of the ultrasound carrier signal (carrier reduction).

2. Process and device according to Claim 1, characterized in that the ultrasound signal is suppressed (turned mute) during modulation pauses, in other words, when no audio signal is to be reproduced.

3. Process and device according to one of the above Claims, characterized in that the audio signal that is to be reproduced (and that is still a low-frequency signal) is subjected to frequency response linearization prior to modulation.

4. Process and device according to one of the above Claims, characterized in that the audio signal, which is to be reproduced, is subjected to a two-sideband amplitude modulation or a one-sideband amplitude modulation.

5. Process and device according to one of the above Claims, characterized in that means are provided for suppressing the ultrasound carrier by an amount of about 8 to 20 dB, preferably 12dB.

6. Process and device according to one of the above Claims, characterized in that the frequency of the ultrasound carrier signal is in the range of about 40kHz to 500kHz.

7. Process and device according to one of the above Claims, characterized in that, in the course of a two-sideband amplitude modulation, means are provided for suppressing the lower sideband.

8. Process and device according to one of the above Claims, characterized in that means are provided for performing a rectification (frequency response linearization) after amplitude modulation.

9. Process and device according to one of the above Claims, characterized in that there is provided a plurality of ultrasound converters that are connected parallel.

10. Process and device according to Claim 9, characterized in that the converters are arranged as tightly together as possible on a plate.

11. Process and device according to one of the above Claims, characterized in that the modulation is performed by means of a digital signal processor.

12. Process and device according to one of the above Claim, characterized in that a water-airbubble mixture is arranged in the ultrasound propagation path.

13. Process and device according to Claim 12, characterized in that the water-air-bubble mixture is made in an earphone ear piece.

14. Process and device according to one of the above Claims, characterized in that—within the propagation path of the ultrasound beams—there is arranged a sound-range permeable medium that contains cavities which, together with the media material, display a plurality of Helmholz-resonators that are preferably attuned to the first blood pressure, pulse rate upper harmonic wave of the ultrasound signal.

15. Process and device according to Claim 14, characterized in that the cavities are filled with a nonlinear medium.

16. Process and device according to one of the above Claims, characterized in that a plurality of ultrasound converters are arranged in ring fashion.

17. Process and device according to one of the above Claims, characterized in that the ultrasound carrier sound signal and the sideband signal are supplied to separate converters.

18. Process and device according to one of the above Claims, characterized that the angle of opening of an ultrasound converter is roughly in the range of 0.5 to 10°, preferably 1°.

19. Process and device according to one of the above Claims, characterized in that means are provided for subjecting the audio signal to a pre-distortion.

20. Process and device according to one of the above Claims, characterized in that means are fashioned for swinging the ultrasound beam into a desired direction.

21. Process and device according to Claim 20, [characterized in] that means [are provided] for swinging the ultrasound beam, [consisting] of a mechanical swinging device of the ultrasound radiator and/or [consisting] of an electronic triggering of the ultrasound radiators in the manner of a so-called "phased array" and/or that a swingable reflector is made which will reflect the ultrasound in a desired direction.

22. Process and device according to one of the above Claims, characterized in that the ultrasound device forms an ultrasound wallpaper so that, when listening in, the

impression is created that the sound comes directly from the wall (or the wallpaper on the wall).

23. Process and device according to one of the above Claims, characterized in that the carrier ban of the ultrasound beam ban and the ultrasound beam sideband are generated with different converters.

24. Process and device according to one of the above Claims, characterized in that the audio-low-frequency signal is subjected to a psychoacoustic pre-processing (in particular, a psychoacoustic pre-distortion) and that appropriate means are provided for this purpose.

25. Process and device according to one of the above Claims, characterized in that the device is made as acoustic tape so that only the moved listener is exposed to sound waves—but not the surrounding area—when a listener moves past an ultrasound converter.

26. Process and device according to one of the above Claims, characterized in that the is provided at least one ultrasound converter that is used exclusively or additionally for ultrasound radiation as transmitter device and/or as receiver device of a distance measurement instrument based on ultrasound.

27. Process and device according to one of the above Claims, characterized in that the properties of the audio signal, that is to be reproduced, in particular the audio signal whose lower boundary frequency is determined by the size of the reflection surface, in order thus to preferably compensate the frequency response linerization or the rectification of the audio signal.

28. Process and device according to one of the above Claims, characterized in that the audio signal to be reproduced is subjected, in a modulator, to a frequency modulation and/or phase modulation.

29. Use of an ultrasound reproduction device according to one of the above Claims in an art exhibit and/or in a museum or for active noise compensation and/or in conference systems, and/or as loudspeaker, as earphone replacement and/or for the specifically directed exposure to sonic waves on a stage (prompter) and/or in the form of addressable loudspeakers and/or for the purpose of exposing computer workstations to soundwaves and/or as Surround loudspeaker and/or for the acoustic exposure to sonic waves of certain specific zones and/or in a no-hands system.

30. Use according to one of the above Claims, for exposing, to sonic waves, an area through which the listener is moving or through which the listener is being moved, while the reproduction level of the ultrasound signal is always aimed at the moved listener.

[see keyed source for Figure 1, page 1/8]

Figure 1. Simple implementation of ultrasound loudspeaker

[key:] 1—low frequency; 2—frequency response linerization 1; 3—two-sideband amplitude modulation, carrier in ultrasound range; 4—error compensation; 5—frequency response linerization 2

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[see keyed source for Figure 2, page 2/8]

Figure 2. Improved circuitry of ultrasound loudspeaker

[key:] 1—low frequency; 2—frequency response linearization 1; 3—one-sideband amplitude modulation, carrier in ultrasound range and suppressed by about 12dB; 4—error compensation; 5—frequency response linearization 2

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[see keyed source for Figure 3, page 3/8]

Figure 3. Transmission path made with digital signal processing

[key:] 1—low frequency; 2—digital signal processing

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[see keyed source for Figure 4, page 4/8]

Figure 4a

[key:] 1—intermodulation products; 2—$2f_1 + \Delta f$ (inaudible); 3—Δf (audible);
4—f_1 = carrier frequency (ultrasound with constant amplitude) Δf=audiosignal

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[see keyed source for Figure 4b, page 5/8]

Figure 4b

[key:] 1—audio sound wave 1kHz; 2—ultrasound converter

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[see keyed source for Figure 5, page 6/8]

Figure 5

[key:] 1—resultant directional characteristics; 2—speed of sound

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[see keyed source for Figure 6a, page 7/8]

Figure 6a

[key:] 1—object (picture, etc.); 2—imaginary sound source; 3—the sound is perceived by the listener exclusively from the direction of the object

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[see keyed source for Figure 6b, page 8/8]

Figure 6b

[key:] 1—object 1 (picture, etc.); 2—object 2 (picture, etc.); 3—differing data that do not influence each other

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1/8

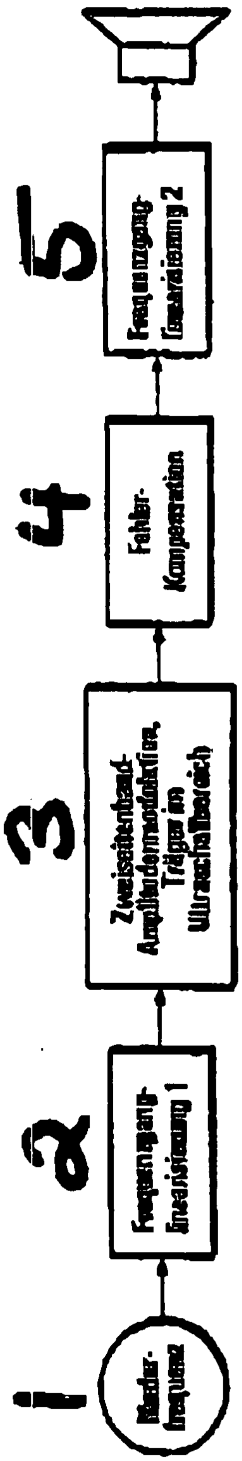

Fig. 1 Einfache Realisierung des Ultraschall - Lautsprechers

2/8

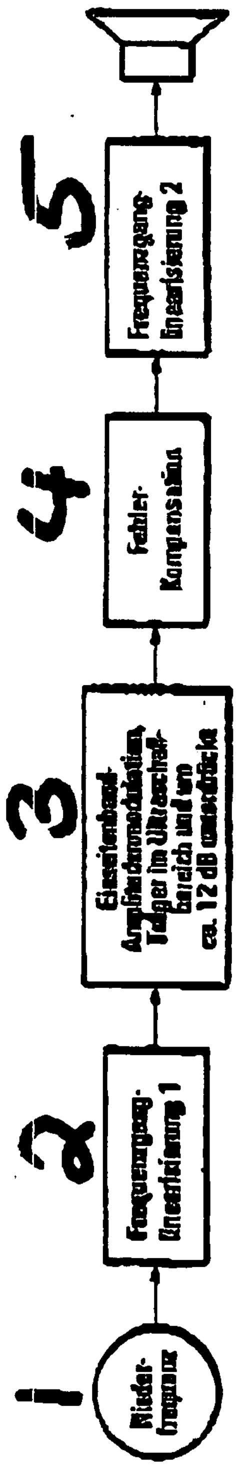

Fig. 2 Verbesserte Schaltung des Ultraschall - Lautsprechers

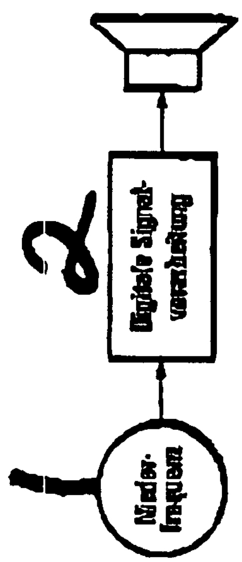

Fig. 3 Übertragungsstrecke realisiert mit digitaler Signalverarbeitung (DSP)

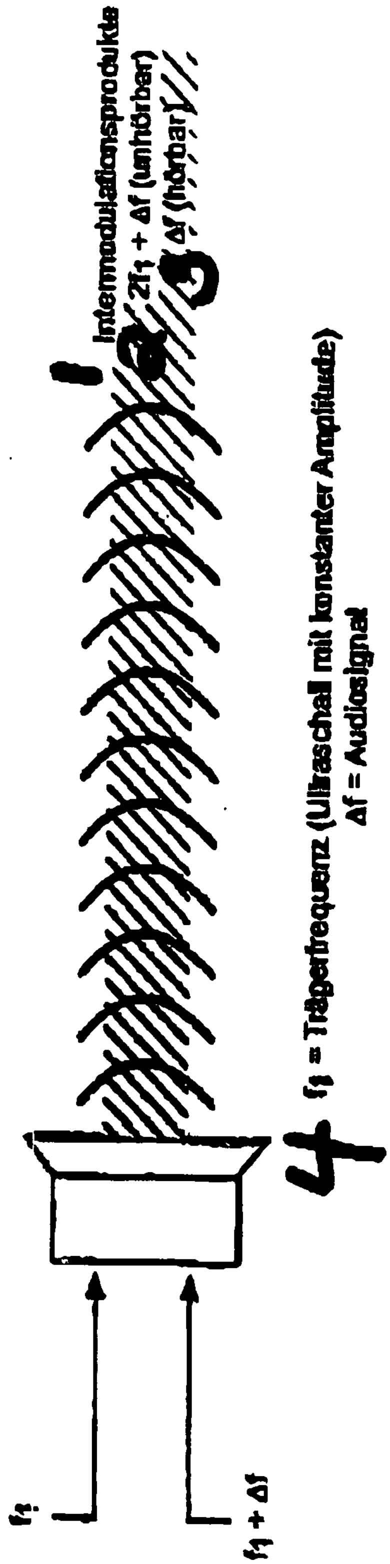
Intermodulationsprodukte
$2f_1 + \Delta f$ (unhörbar)
Δf (hörbar)
f_1 = Trägerfrequenz (Ultraschall mit konstanter Amplitude)
Δf = Audiosignal
f_1
$f_1 + \Delta f$
1
2
3
4

Fig. 4a

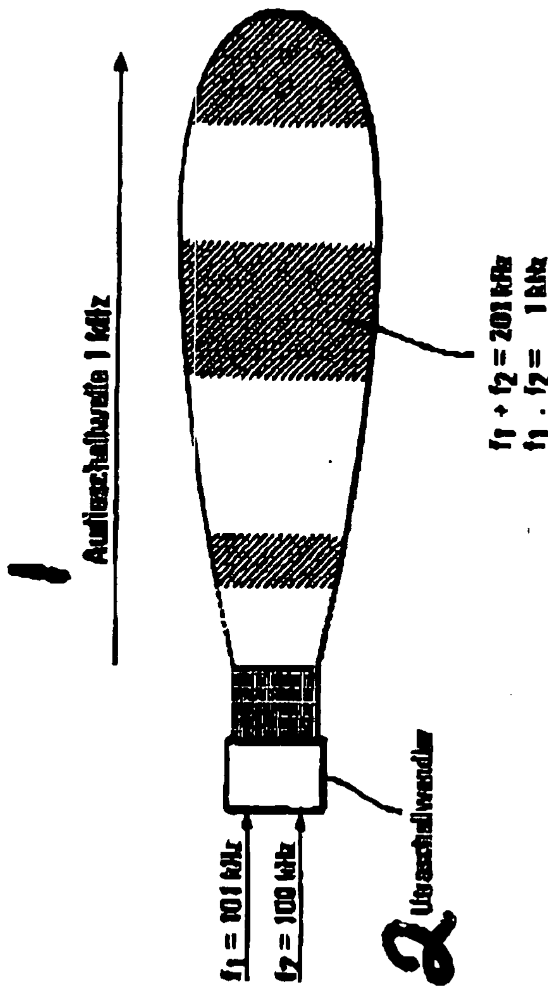

Fig. 4b

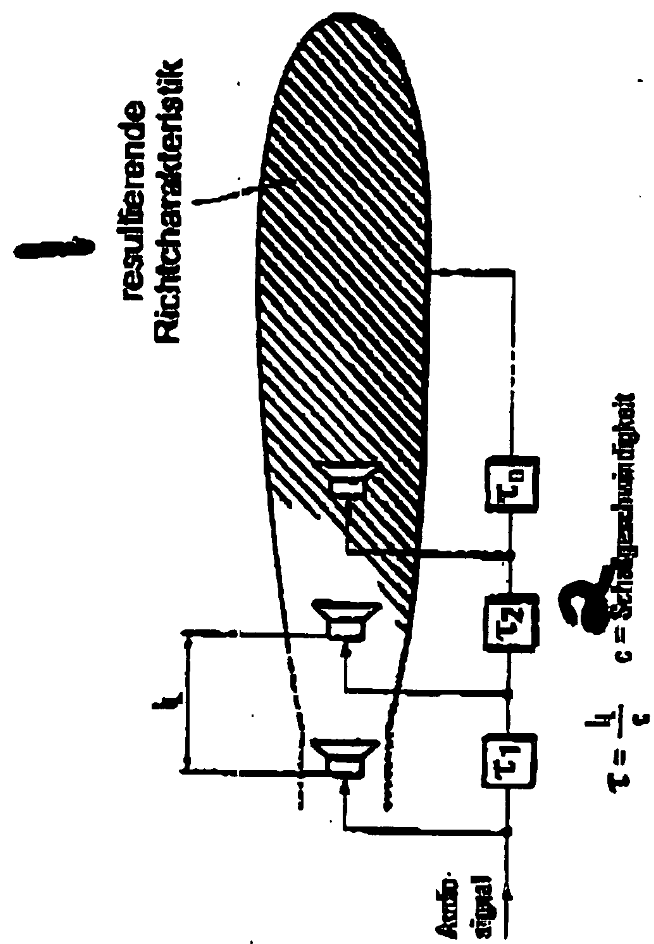
resultierende
Richtcharakteristik
L
τ_1
τ_2
τ_n
Audio-
signal
$\tau = \frac{L}{c}$
c = Schallgeschwindigkeit

Fig. 5

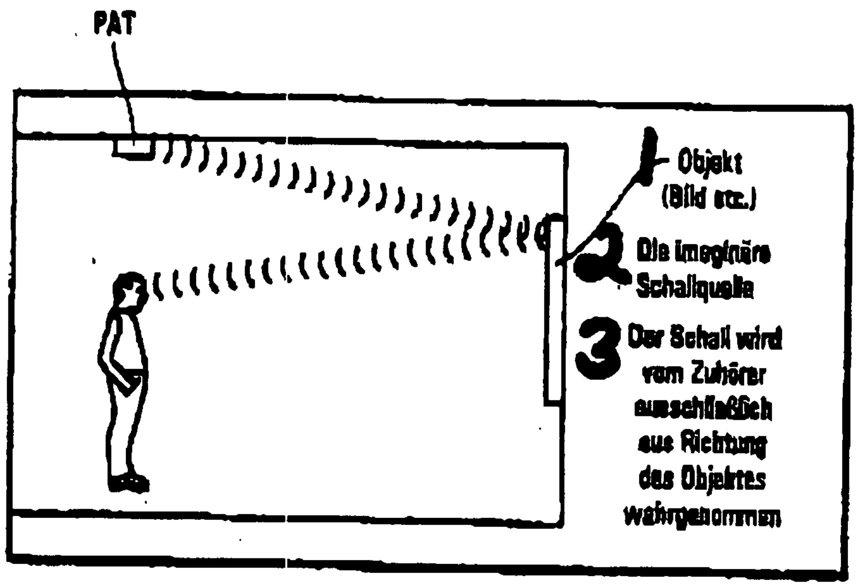
PAT
1 Objekt (Bild etc.)
2 Die imaginäre Schallquelle
3 Der Schall wird vom Zuhörer ausschließlich aus Richtung des Objektes wahrgenommen

Fig. 6a

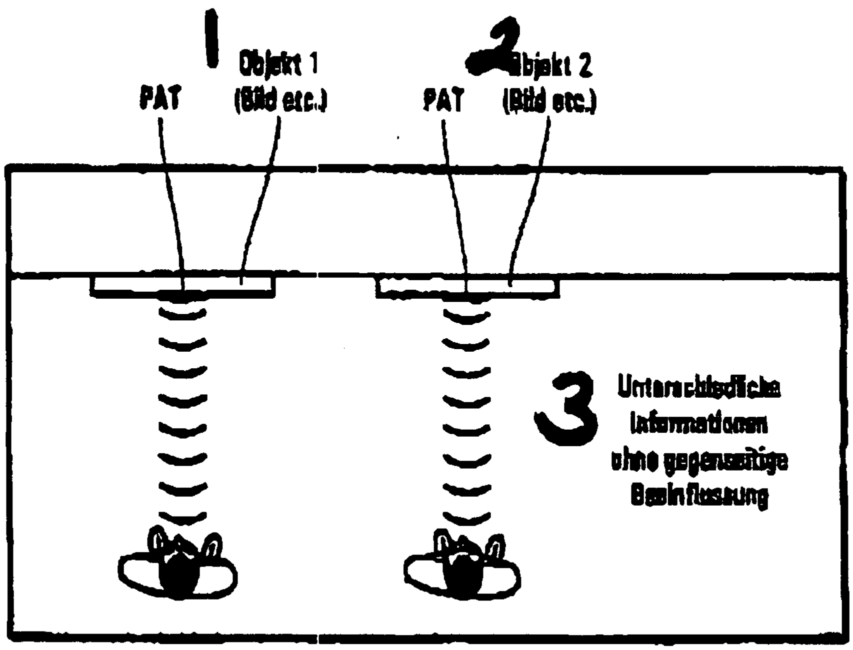
1
Objekt 1
PAT
(Bild etc.)
2
Objekt 2
PAT
(Bild etc.)
3
Unterschiedliche
Informationen
ohne gegenseitige
Beeinflussung

Fig. 6b